Release Note

SIP CLIENT V2.23

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Firmware name:	SIP CLIENT V2.23
FW Release Date:	10. Nov. 2022
Package Name:	abcl_sip_v2.23_20221110.zip
Version Overview:	**Type** / **File name** / **Version Date** Firmware / abclw.rom / VB1.23_20220303 WEB UI / abclapp.cob / V02.07_20140820 FW Ext. 1 / sg.bin / V10.18_20171019 FW Ext. 2 / fs.bin / V02.08_20140820 FW Ext. 3 / bclio.bin / V02.06_20151028 Bootloader / exfull.rom / V99.28_20160624

Type	File name	Version Date
Firmware	abclw.rom	VB1.23_20220303
WEB UI	abclapp.cob	V02.07_20140820
FW Ext. 1	sg.bin	V10.18_20171019
FW Ext. 2	fs.bin	V02.08_20140820
FW Ext. 3	bclio.bin	V02.06_20151028
Bootloader	exfull.rom	V99.28_20160624

1 Hardware Compatibility

This firmware is fully supported on the following devices:

- Annuncicom PS1, PS16, 60, 100, 155, 200, 1000
- Extreamer 100, 105, 110, 120, 200, 205, P5 (all as receivers only), 500
- Instreamer as encoder only

The firmware is partially supported on the following modules:

- IPAM 100/102
- IPAM Carrier Board
- IPAM 300/302

Note: The firmware can run on any other Barix audio devices (e.g. Exstreamer Digital), however the functionality is not guaranteed.

2 Fixed Bugs

The following bugs have been fixed compared to the previously released V2.20 :

- ASIP-280: Wrong values in the Status Report page
- ASIP-281: Notification messaging does not work when the device is not registered
- ASIP-283: No “180 Ringing” sent after reboot in P2P mode
- ASIP-284: SIP Paging on Exstreamer not working
- ASIP-286: Wrong "Call on Level ID" in Status report
- ASIP-290: Remote P2P port wrongly detected when user ID is used in the SIP URI
- ASIP-292: favicon_barix.ico is missing
- ASIP-293: Full duplex device wrongly detected as receive only
- ASIP-295: applications.sh does not work on 64 bit MacOS
- ASIP-296: Delayed call setup with 3CX server

3 New Features and Improvements

The following new features have been implemented compared to the previously released V2.20:

- Make the listen SIP and RTP ports configurable also in P2P mode
- ASIP-287: Use Ann60 digital input as offhook/onhook button

4 Known Issues

1. When a X8 contact that is configured as input is tried to be set using DTMF command 3, the active call will be closed as it will be detected as a button press. Calculate properly the bitmask used in this command so that it does not try to set to “1” X8 pins configured as inputs. This issue is valid also for all versions of SIP client prior to V2.01;

2. In order the status of X8 contacts to be correctly displayed after using DTMF command 4, the SIP client needs to be rebooted in order the X8 device to be properly reinitialised with the new settings.

3. The inputs supervision is fully supported only on Annuncicom 155. When run on Annuncicom 1000, the inputs are shown as normal 2 state (open/close) inputs.

4. Decoding DTMF commands does not work properly when the remote peer is sending them both as DTMF digits and via the SIP INFO message. In this case configure the remote peer to use only one of these methods.

5. The home page of the SIP client being transferred displays the remote ID of the transferee instead of the ID of the party to which the call has been transferred to. I.e, if SIP Client A calls B, and B transfers the call to C, the home page of SIP Client A will still show the remote ID of B instead of C after the transfer. This is because SIP client A does not get any information that it is being transferred to C-just that it is put on hold temporary while being transferred. All the call handling in the case of Blind Call transfer is done by the SIP server and therefore “invisible” for SIP client A.

6. The implementation of “***ASIP-287: Use Ann60 digital input as offhook/onhook button***” feature mimics the behaviour of a standard Analog Telephone-it dials/answers a call then the handset goes off-hook (digital input 0 is pressed), and closes the call when the handset goes on-hook (digital input 0 is released). For now this feature is intended to be used only with Annuncicom 60 device in “SIP Phone” profile.

   The feature is activated by setting the ***Advanced*** → ***Outbound Calls*** → ***Enable Analog Phone Interface*** option to “On”

   NOTE: This option is visible on the webUI of Annuncicom 60 only

7. When the SIP Client is use as base for own developments, then be carefully. The current application is already using much resources, there is not much space left. Optimally you remove not required code/functionality before you add code.

5 Release generating details[1]

5.1 GIT release tag

Checkout the barix and bcl GIT repositories with the following tag:

abcl_sip_v2.23_20221110

5.2 Compile commands:

Run the following command from the cloned barix folder:

```
make ab_sip
```

To generate the combined FW image that contains both the SIP FW and the Annuncicom IC FW (that is used in the production process) use the command:

```
make mu_sipann
```

5.3 Regenerating the FW image

The source code of the SIP Client application is too big in size, and cannot fit into the COB file together with the tokenized BCL code. The generated release already has the source code stripped out of the FW image. However, if for some reason manual rebuild of the FW is needed, then the *applications.sh* script or the *applictns.bat* batch file has to be run with the ***-no_source*** option to regenerate the applications.cob file:

```
./applications.sh -no_source
```

1 This information is intended for BARIX developers only

6 Legal Information

For information about our devices and the latest version of this manual please visit www.barix.com

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